4800VS-3

Call Station Software

User Guide

SENTINEL™
EMERGENCY COMMUNICATIONS

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800-558-8957

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Product Overview

The ***4800VS-3*** Call Station is designed to be used only with Cornell's *Sentinel AOR System.* It is a PoE SIP voice intercom device utilized only to activate emergency calls/requests for help and allows for full-duplex 2-way communications between a Base Station Phone and/or a remote monitoring location.

Requires PoE power be provided via a CAT5E or better cable. Maximum cable distance of 100m/328ft applies.

Programming the ***4800VS-3*** is completely done using the Sentinel AOR System Dashboard / programming interface.

Product Specifications

- Physical Faceplate: 6.37" x 4.49**"** (3 Gang); Satin-finished aluminum
- Power Requirements: PoE Only (draws .05A / 50mA standby | .10A / 100mA active)
- Operating Environment: -4°F ~ 122°F
- Speaker Components: Ø 40 mm x 12.7 mm (1.6 x 0.5 in)
- Sensitivity: 90±3dB
- Distortion: <10%, 1000Hz 0.1w/1m
- Rated Power: 3W
- Max Power: 4W
- Frequency Range: 630Hz~4.5KHz
- Acoustics: Mono
- Microphone: -36±2dB RL=2.2K Vs=2.0V
- Relay Switch: Max voltage AC 125V-1A/DC 60V-1A

- Call Button Momentary (normally open) Mushroom button; push-to-call with LED Indicator
- Mounting: Mounting Flush mount in 3-Gang box (min. 2.5" deep)
- Wiring: Cat5e (or better) Ethernet cabling is required for connections between Main System, Base Station(s), Expansion Module(s) and Call Stations. Maximum cable length of 100m/328ft when using premium wire and doing a clean installation. Interference from other cables and building construction may lower this distance.

Basic Settings

General Information

By default, the ***4800VS-3*** Call Station device's IP assignment is configured as DHCP. The Sentinel AOR System creates its own network for all ***4800VS-3*** devices to use to interact and communicate together. Cornell pre-programs the call stations to have a Static IP Address on the Sentinel network which can be used to access the device web management interface.

A list of all device IP Address is provided with each system and should be kept nearby. Press and hold the red call button for 5 seconds then release to have a call station announce its IP Address. You can also check the IP Address of each device using the Sentinel Dashboard used to make system programming changes. Contact Cornell's Tech Support team at 800-558-8957 if you need a copy sent to you. Please have the site name or SO# ready to lookup the information.

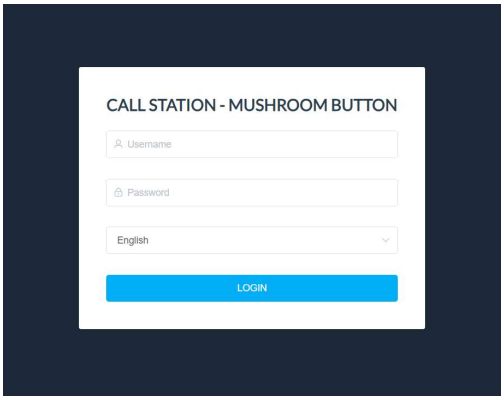
NOTE: There is typically no need to adjust any settings inside the ***4800VS-3*** Call Station devices except audio volumes (see page 9 for details on making volume changes).

Getting Connected

Simply connect a laptop to the Sentinel V2 unit (***SCM-4825***) port labeled "PC" using an ethernet patch cable. Make sure the laptop ethernet port is set to "DHCP" to automatically obtain an IP Address. Open a web browser and enter the IP Address of the ***4800VS-3*** Call Station device on the Sentinel AOR System network to access the device and make modifications to settings.

Web Interface Login

If the steps above have been done correctly, a login page will appear for the "CALL STATION – MUSHROOM BUTTON". Use the default login credentials below:

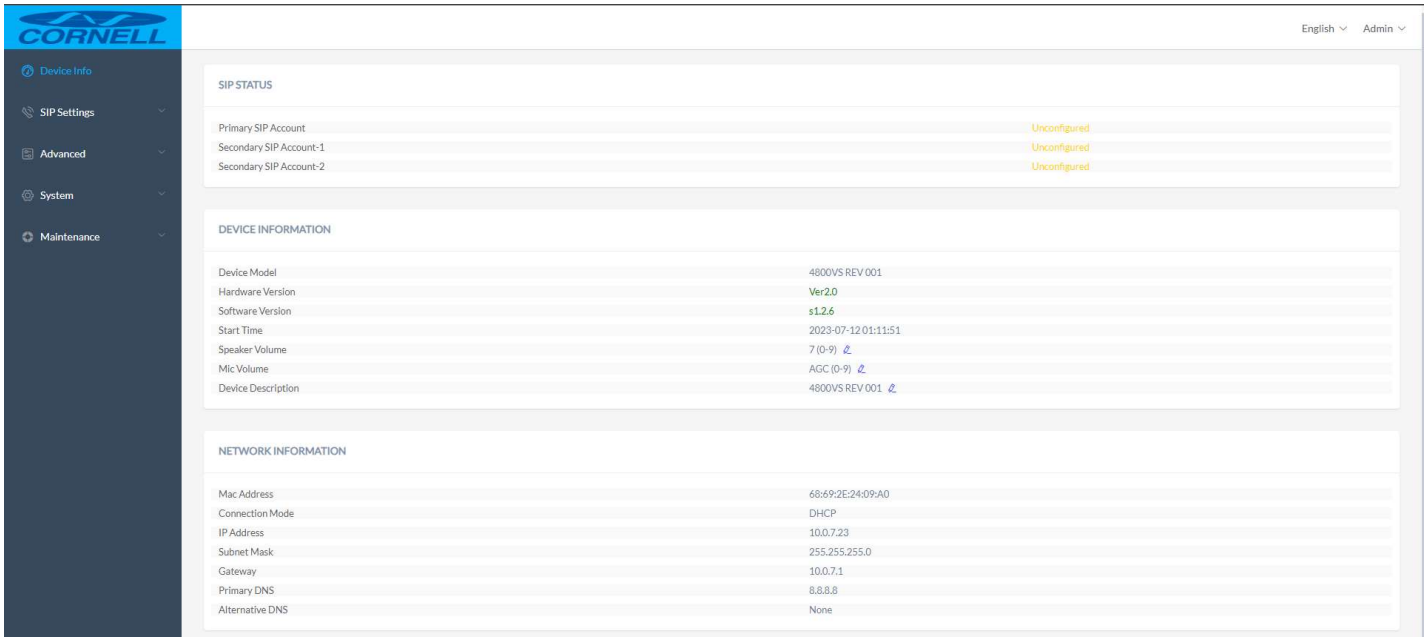


Username: admin
Password: admin

Select the **LOGIN** button to proceed.

Device Info

Upon logging into the device, you will be presented with general device information as shown below:



- **SIP STATUS**
  - Primary and Secondary general SIP account information and status

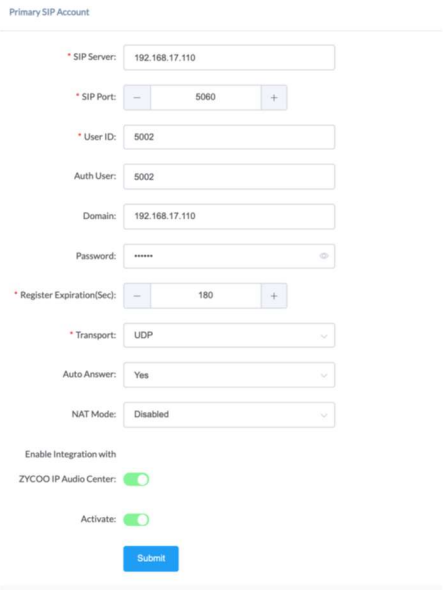
- **DEVICE INFORMATION**:
  - **Device Model**: device model # of the device (4800VS-3) and
  - **Hardware Version:** device hardware version
  - **Software Version:** device software version (can be upgraded)
  - **Start Time:** last startup time for the device
  - **Speaker Volume:** current volume level for the device speaker
  - **Mic Volume:** current volume level for the device microphone
  - **Device Description:** this value will be the displayed browser tab name for the device
    - After modification, the tab name will change accordingly

- **NETWORK INFORMATION**
  - **Mac Address:** current MAC Address for the device
  - **Connection Mode:** type of network connection (can be either DHCP or STATIC)
  - **IP Address:** current network IP Address for the device
  - **Subnet Mask:** current subnet mask for the device
  - **Gateway:** current default gateway used for the device
  - **Primary DNS:** current primary DNS for the device
  - **Alternative DNS:** current alternative DNS for the device

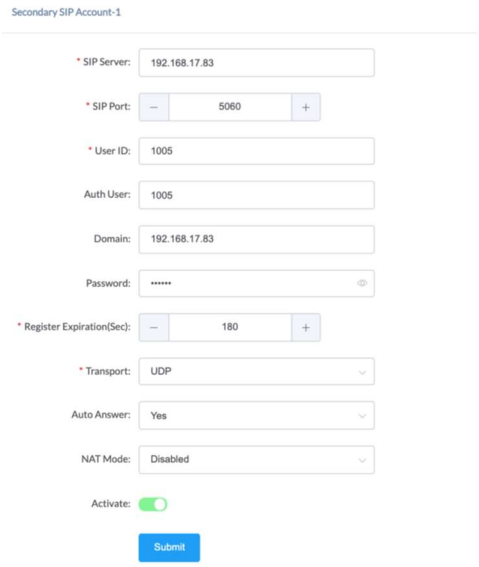
SIP Settings

SIP Accounts

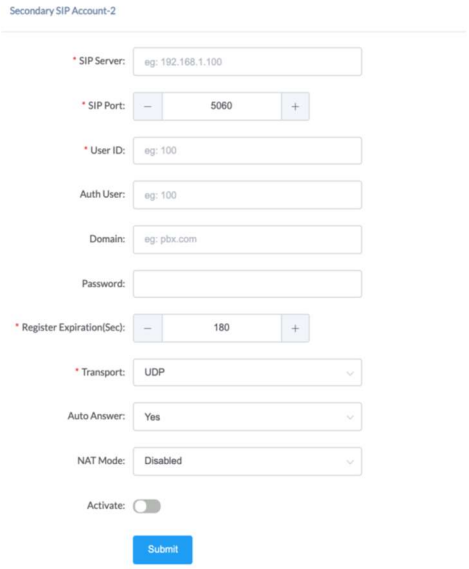
Primary SIP Account



Secondary SIP Account1



Secondary SIP Account2

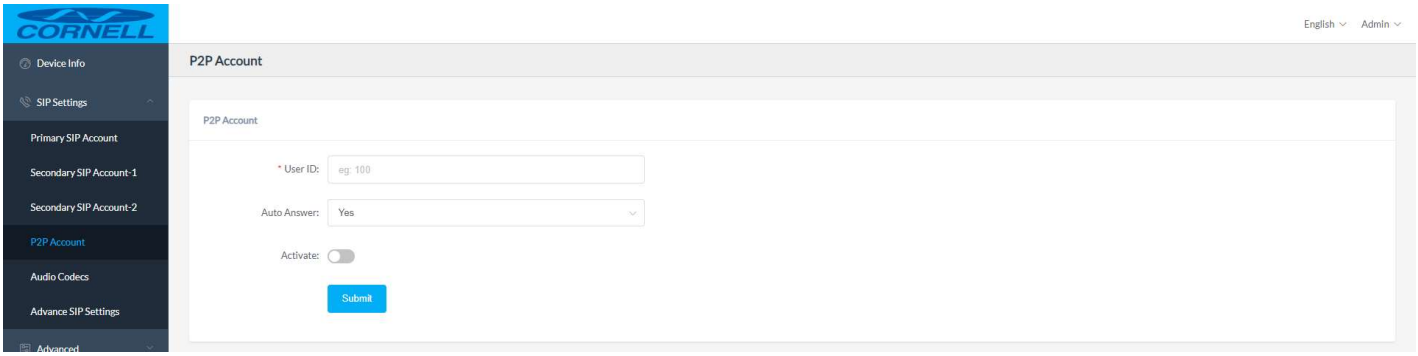


- **Primary SIP Account, Secondary SIP Account options:**
  - **SIP Server:** the IP Address or domain name of the SIP server
  - **SIP Port:** default SIP port is 5060 but can be modified as needed
  - **User ID:** the SIP account number provided by the SIP server
  - **Auth User:** username for the authorized SIP account
  - **Domain:** SIP domain name
  - **Password:** the authorized SIP account password
  - **Register Expiration:** SIP register expiration time (default is 180 seconds)
  - **Transport:** the transport protocol used (choose from UDP, TCP, TLS)
  - **Auto Answer:** default is Yes option (choose from Yes, No, Answer Delay)
    - Ring Tone: ring tone played before call is answered (when Auto Answer is NO)
    - Answer Delay: set the amount of time the ring tone plays before call is auto answered
  - **NAT Mode:** set the NAT mode and options (choose from STUN, TURN, or ICE)
  - **Enable Integration with Zycoo IP Audio Center:**
    - Enable/Disable if device needs to interact with the Audio Center (disabled by default)
  - **Activate:** Enable/Disable the SIP register feature for the device

*****IMPORTANT!*****

All of the above settings are automatically configured during the programming process when using the Sentinel AOR System dashboard / programming interface and should not be adjusted. Contact Cornell Communications at 800-558-8957 before making any changes to these device settings.

P2P Account



- **P2P Account**
  - **User ID:** name displayed as the outgoing number when device calls out
  - **Auto Answer:** default is Yes option (choose from Yes, No, Answer Delay)
    - Ring Tone: ring tone played before call is answered (when Auto Answer is NO)
    - Answer Delay: set the amount of time the ring tone plays before call is auto answered
  - **Activate:** Enable/Disable the P2P feature

Audio Codecs

- **Audio Codecs**
  - **G722:** Enable/Disable this codec
  - **G.711(Ulaw):** Enable/Disable this codec
  - **G.711(Alaw):** Enable/Disable this codec
  - **Opus:** Enable/Disable this codec

Note: Keep at least 1 codec activated that is supported by the SIP server or SIP paging will not work.

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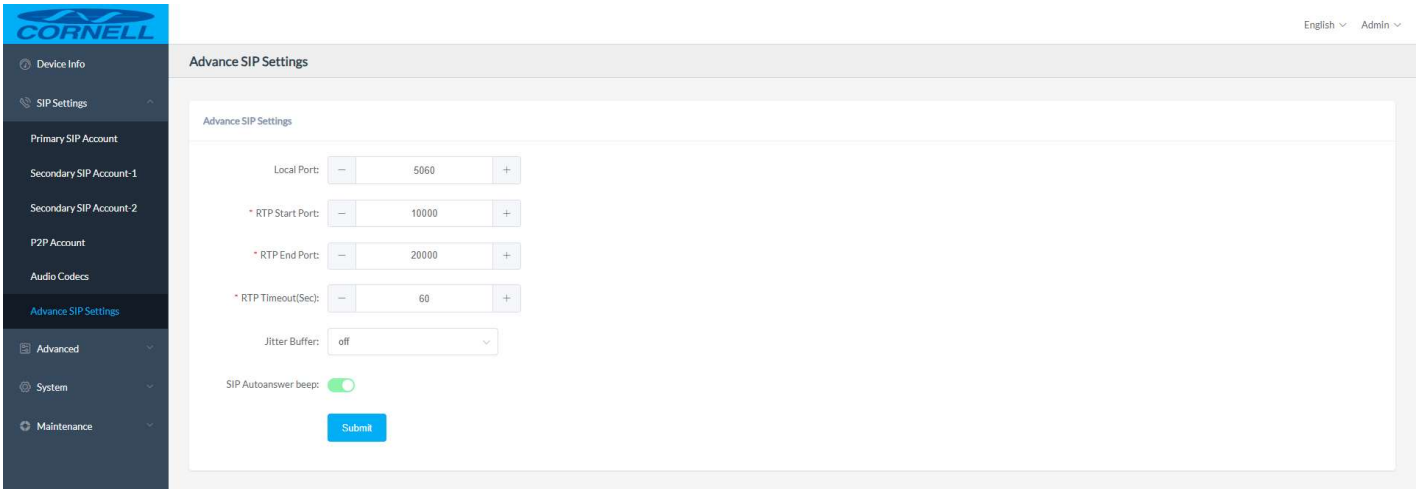
All of the above settings are automatically configured during the programming process when using the Sentinel AOR System dashboard / programming interface and should not be adjusted. Contact Cornell Communications at 800-558-8957 before making any changes to these device settings.

Advanced SIP Settings



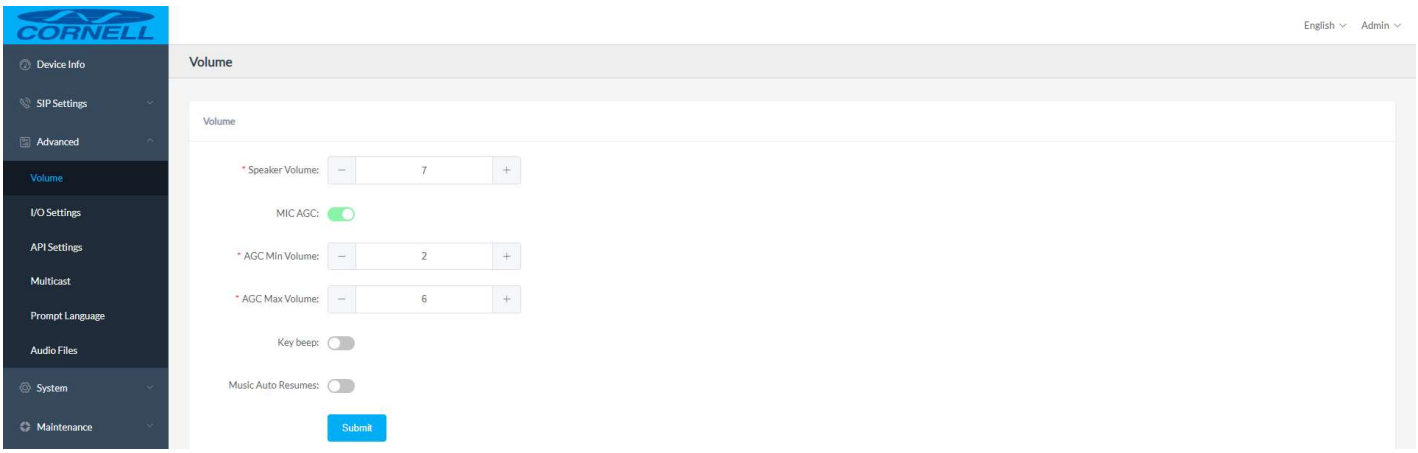
- **Advance SIP Settings**
  - **Local Port:** the port used to receive SIP packets
  - **RTP Start Port:** the starting RTP port used for media sessions
  - **RTP End Port:** the end RTP port used for media sessions
  - **RTP Timeout(Sec):** time range (in seconds) that if the system does not receive the RTP stream, then the call will end
  - **Jitter Buffer:** represents the Jitter Buffer where voice packets are collected, stored, then sent to the voice processor in even intervals to improve audio quality (choose from off, adaptive, fixed)
    - **Off:** disabled
    - **Adaptive:** can adjust Jitter Buffer based on delays in the network
    - **Fixed:** adds a fixed delay to voice packets
  - **SIP Autoanswer beep**: Enable/Disable the ringtone beep when a call comes (only applies when the SIP Autoanswer feature is enabled)

*****IMPORTANT!*****

All of the above settings are automatically configured during the programming process when using the Sentinel AOR System dashboard / programming interface and should not be adjusted. Contact Cornell Communications at 800-558-8957 before making any changes to these device settings.

Advanced Settings

Volume

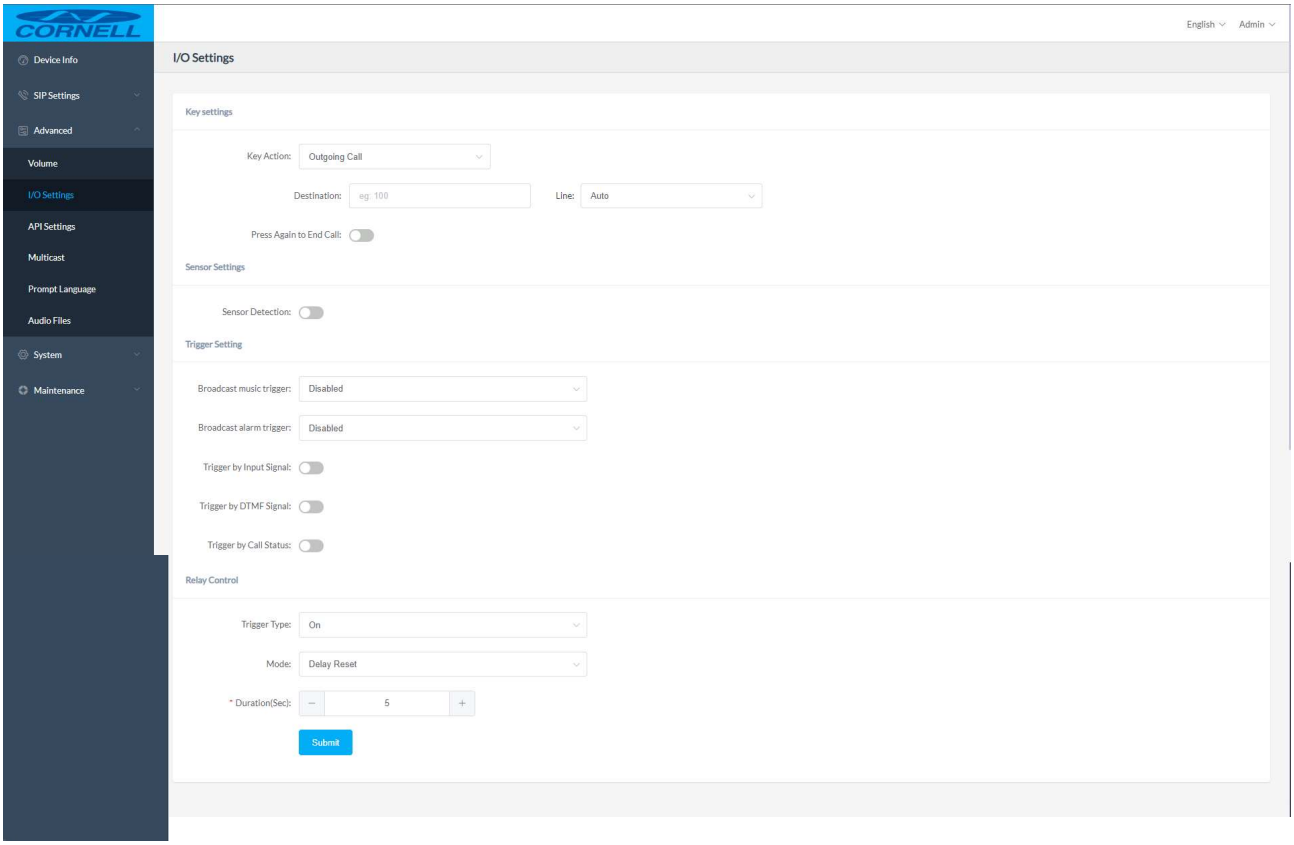


- **Volume**
  - **Speaker Volume:** default is set by Cornell; adjustable range is 0 – 9.
  - **MIC AGC:** When enabled (default), the system will automatically adjust the microphone volume according to the environment. User can only adjust mic volume when this setting is disabled.
  - **AGC Min Volume:** represents the minimum value of the automatic gain control.
  - **AGC Max Volume:** represents the maximum value of the automatic gain control.
  - **Key Beep:** Enable/Disable the beep sound from the key button.
  - **Music Auto Resumes**: Enable/Disable the music to automatically resume when used.

*****IMPORTANT!*****

All of the above settings are automatically configured during the programming process for optimal performance with the Sentinel AOR System. Adjust volumes in increments of 1 or 2 at most, then test the changes. Contact Cornell Communications at 800-558-8957 with questions on the Volume adjustments.

I/O Settings



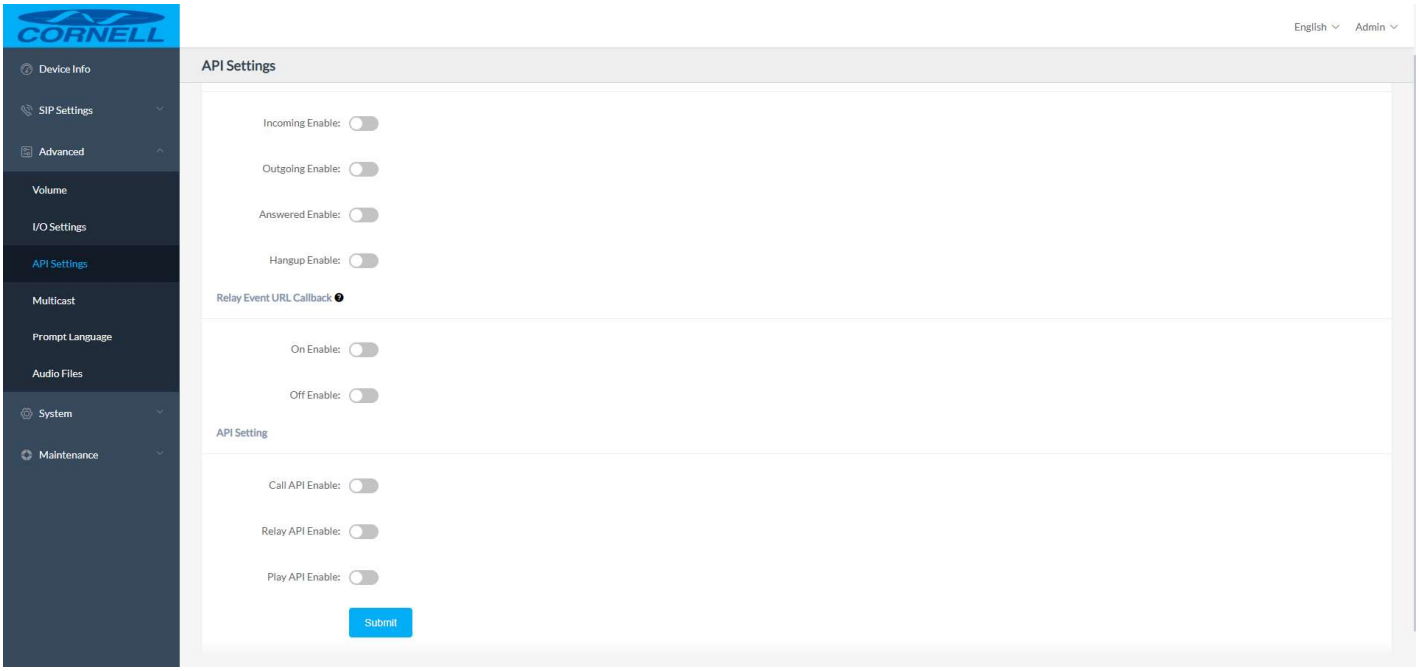
- **Key Settings**
  - **Key Action:** action taken when button is pressed (default is Outgoing Call)
    - **Destination:** the extension or phone number dialed the button is pressed
    - **Line:** the SIP Account or line used to make the Outgoing Call
  - **Press Again To End Call:** Enable/Disable this feature (default is Disabled)

- **Sensor Settings**
  - **Sensor Detection:** Enable/Disable this feature (default is Disabled)

- **Trigger Settings**
  - **Broadcast Music Trigger:** Enable/Disable this feature (default is Disabled)
  - **Broadcast Alarm Trigger:** Enable/Disable this feature (default is Disabled)
  - **Trigger by Input Signal:** Enable/Disable this when using Input signal to trigger relay output
  - **Trigger by DTMF Signal:** Enable/Disable this when using DTMF signal to trigger relay output
  - **Trigger by Call Status:** Enable/Disable a change in call station to signal to trigger relay output

- **Relay Control**
  - **Trigger Type:** set the trigger response (choose from ON, Fast Flashing, or Slow Flashing)
  - **Mode:** set the reset mode for the trigger (choose from Delay, Answered, or Hang-up reset)
  - **Duration(Sec.):** set the Delay Reset time duration in seconds (must choose Delay Reset above)

API Settings



This page is used to configure the API interface of the device. Through the API interface, you can realize device linkage, call control, relay control, and play sound by using the change status of calls and/or the relay.

When either a call or relay event status change occurs, it will trigger an HTTP GET request to call a URL address. Within the URL address, you may use variables to identify some current information as shown below:

1. **${ip}:** The current IP address of the device
2. **${mac}:** The current MAC address of the device
3. **${number}** : The number of the current call

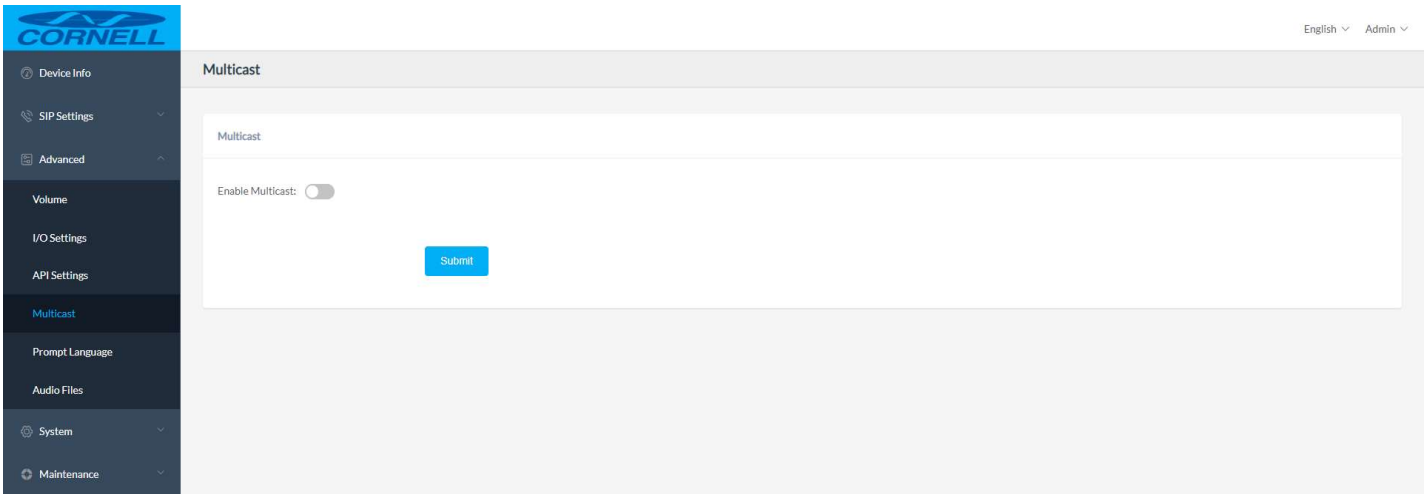
- **Call Event URL Callback**
  - **Incoming Enable:** Enable/Disable incoming calls to trigger API function (default is Disabled)
    - Dry contact relay output triggers when call station device receives an incoming call
  - **Outgoing Enable:** Enable/Disable outgoing calls to trigger API function (default is Disabled)
    - Dry contact relay output triggers when call button is pressed on the call station
  - **Answered Enable:** Enable/Disable answered calls to trigger API function (default is Disabled)
  - **Hangup Enable:** Enable/Disable hangup of calls to trigger API function (default is Disabled)
    - Dry contact relay output triggers when SIP call ends with the call station

- **Relay Event URL Callback**
  - **On Enable:** Enable/Disable Relay ON to trigger API function (default is Disabled)
  - **Off Enable:** Enable/Disable Relay OFF to trigger API function (default is Disabled)

- **API Settings**
  - **Call API Enable:** Enable/Disable Call Events above to trigger API function (default is Disabled)
  - **Relay API Enable:** Enable/Disable Relay Events to trigger API function (default is Disabled)
  - **Play API Enable:** Enable/Disable events to play the API message in HTML (default is Disabled)

Multicast

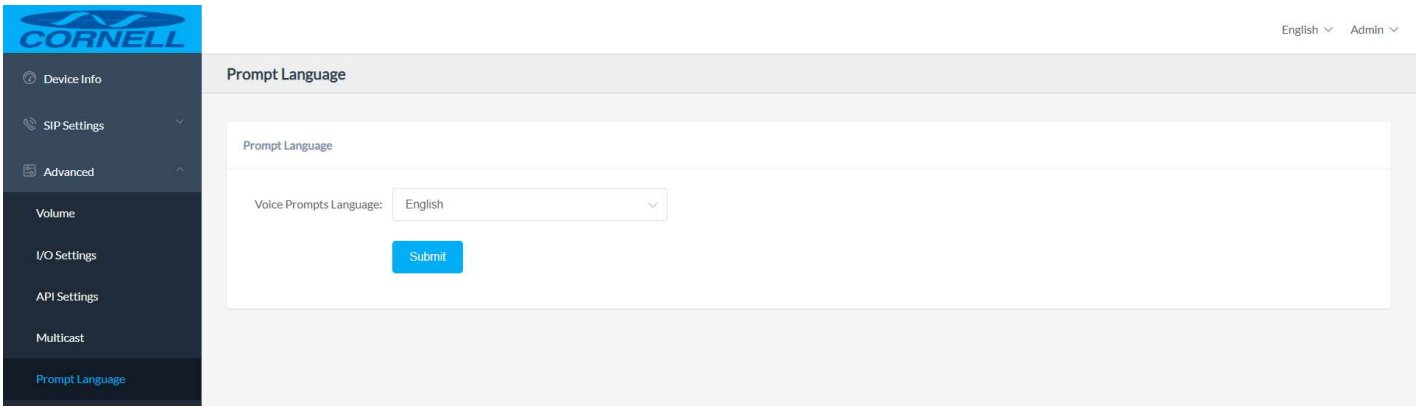
The multicast settings are used to configure the parameters and settings of this function on the 4800VS-3 Call Station device. It can be configured to monitor up to 9 different levels of multicast addresses, the audio streams with a higher priority will interrupt the playback of the lower priority audio streams (disabled by default / function not available).



- **Multicast**
  - **Enable Multicast:** Enable/Disable the Multicast function (default is Disabled)
    - **Priority:** Priority from highest (9) to lowest (1)
    - **Multicast Address:** Address range is 224.0.0.0 – 239.255.255.255
    - **Multicast Port:** Port range is 2000 – 65535
    - **Name:** Create a name for the multicast address
    - **Relay Control:** Options are Disabled, ON, Fast Flashing, or Slow Flashing

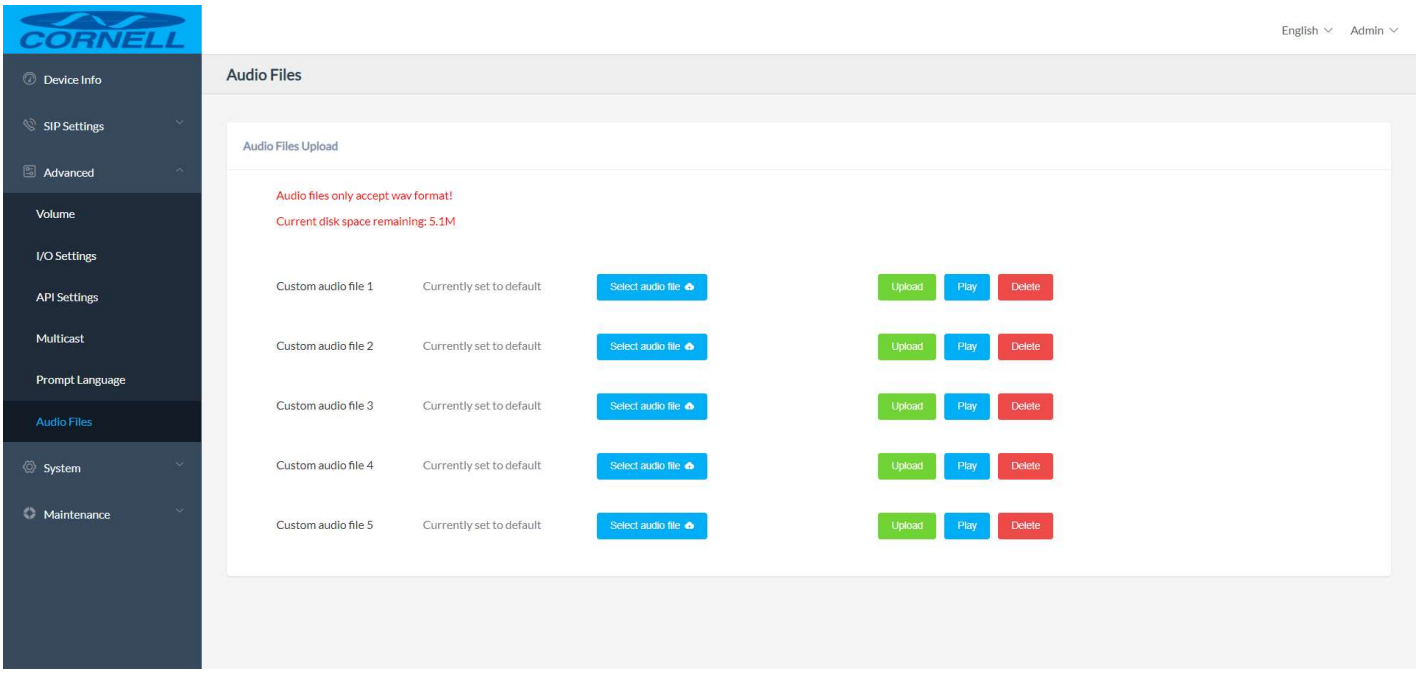
Prompt Language

The language of local voice prompts, like IP address announcements, can be set here.



Audio Files

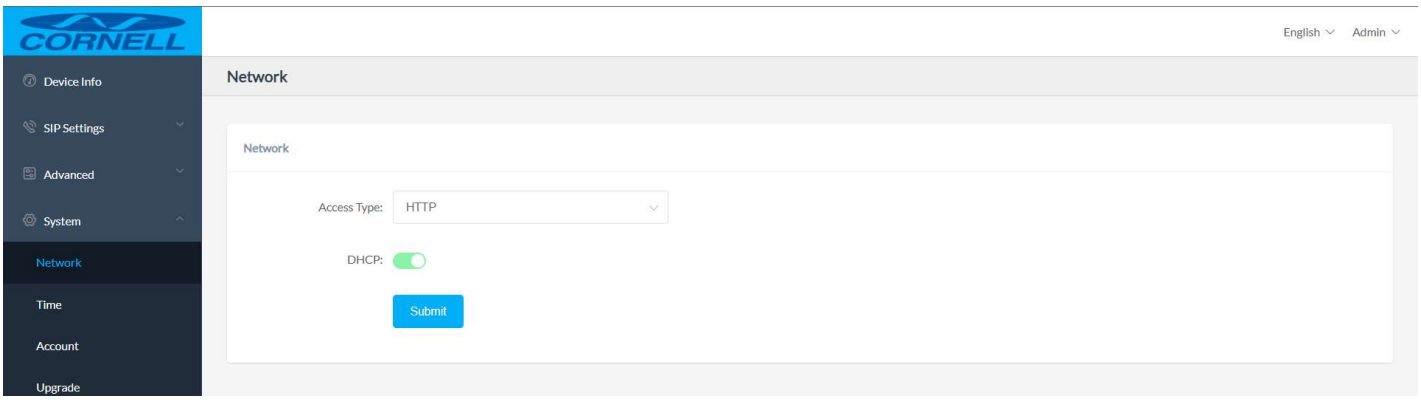
The Audio files section allows users to self-upload up to 5Mb of audio files to the endpoint and use it as ringtone or Play API audio file. Click on the Select Audio File button to choose a local file, then the Upload button to upload the local file. Click on the Play button to test the message or Delete button to remove the audio file.

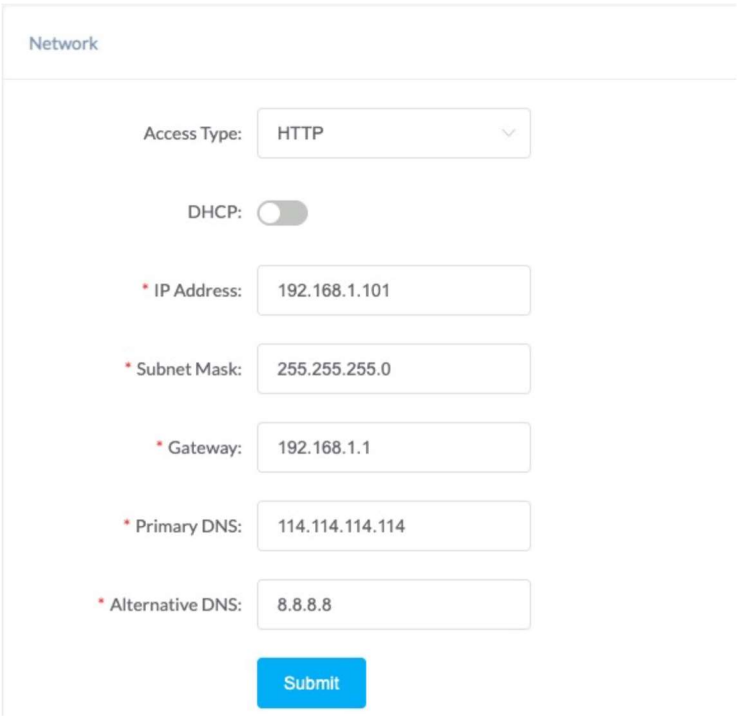


System Settings

Network

- **Network**
  - **Access Type:** Set the access type here (default is HTTP)
  - **DHCP:** Enable/Disable DHCP for the device (shows network settings when Disabled)

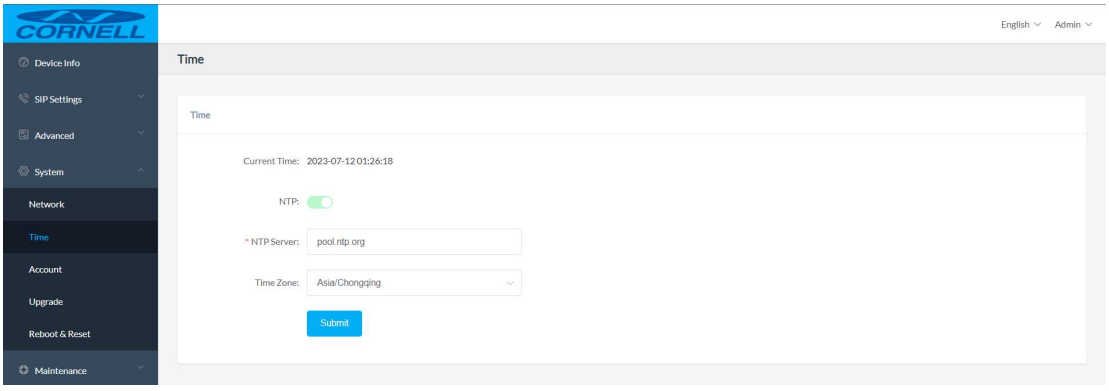




- **Network**
  - **Access Type:** Set the access method of the website (choose from HTTP or HTTPS)
  - **DHCP:** Enable/Disable DHCP for the device (shows network settings when Disabled)
  - **IP Address:** enter a vacant IP Address for the call station device within your LAN
  - **Subnet Mask:** Set the subnet mask of your LAN
  - **Gateway:** Set the Network Gateway IP Address of your LAN
  - **Primary DNS:** Set the Primary DNS server address
  - **Alternative DNS:** Set an Alternative DNS server address (used when Primary DNS fails)

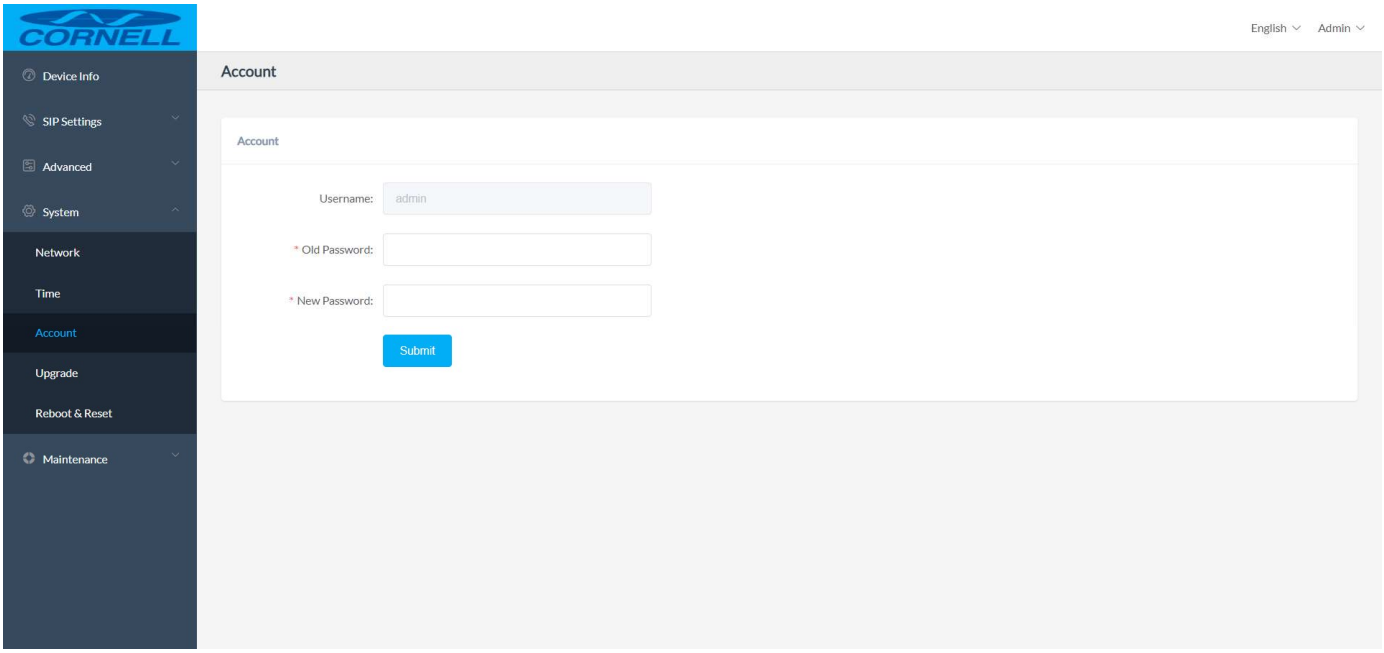
Time

Enable and set the NTP, NTP Server and Time Zone to use for the device (default is Disabled).

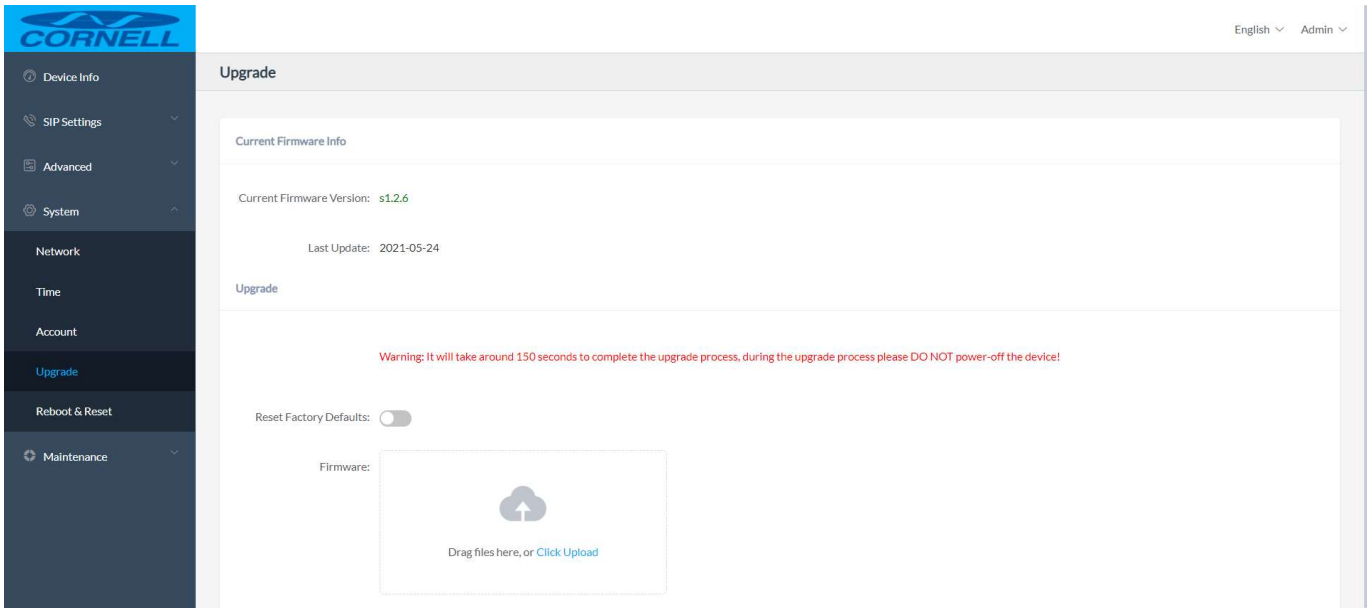


Account

- **Account**
  - **Username:** Set the username used to login to device (default is admin)
  - **Old Password:** Enter the current password used to login to device (default is admin)
  - **New Password:** enter a new password used to login to device

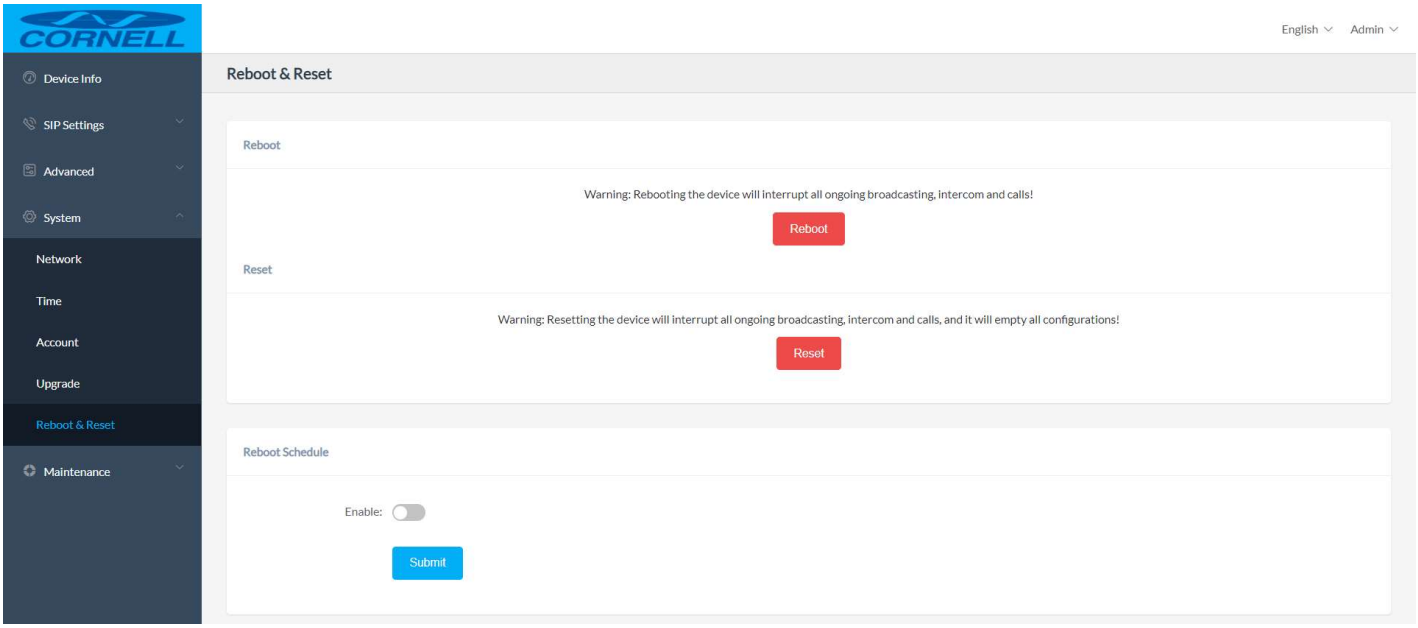


Upgrade



View the current firmware information, upgrade the firmware to another version, or restore factory default settings for the device on this page.

Reboot & Reset

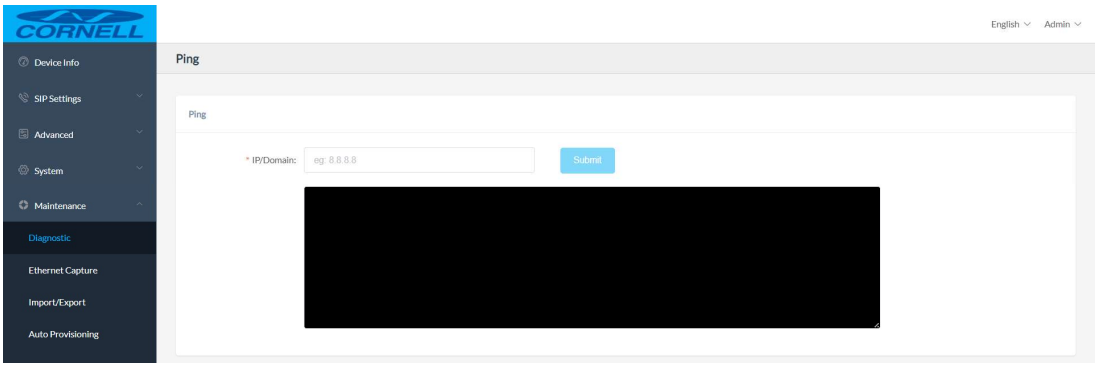


- **Reboot**
  - Reboot the 4800VS-3 call station device
- **Reset**
  - Restore 4800VS-3 call station device to factory default settings
- **Reboot Schedule**
  - Create and set a scheduled reboot cycle (default is Disabled)

System Settings

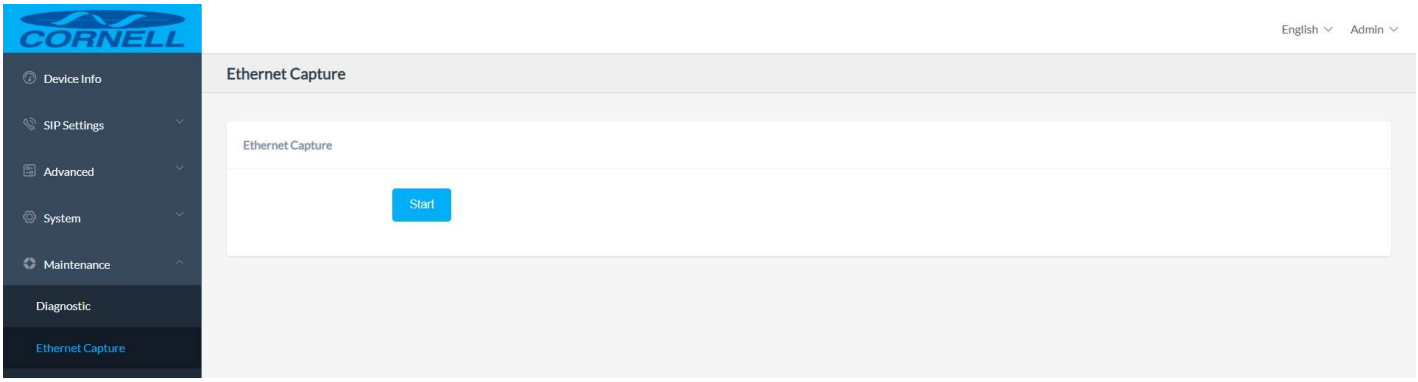
Ping

The ping function is an administrative tool used to check/test network connectivity. Enter a valid IP address and click Submit to view the network trace route.



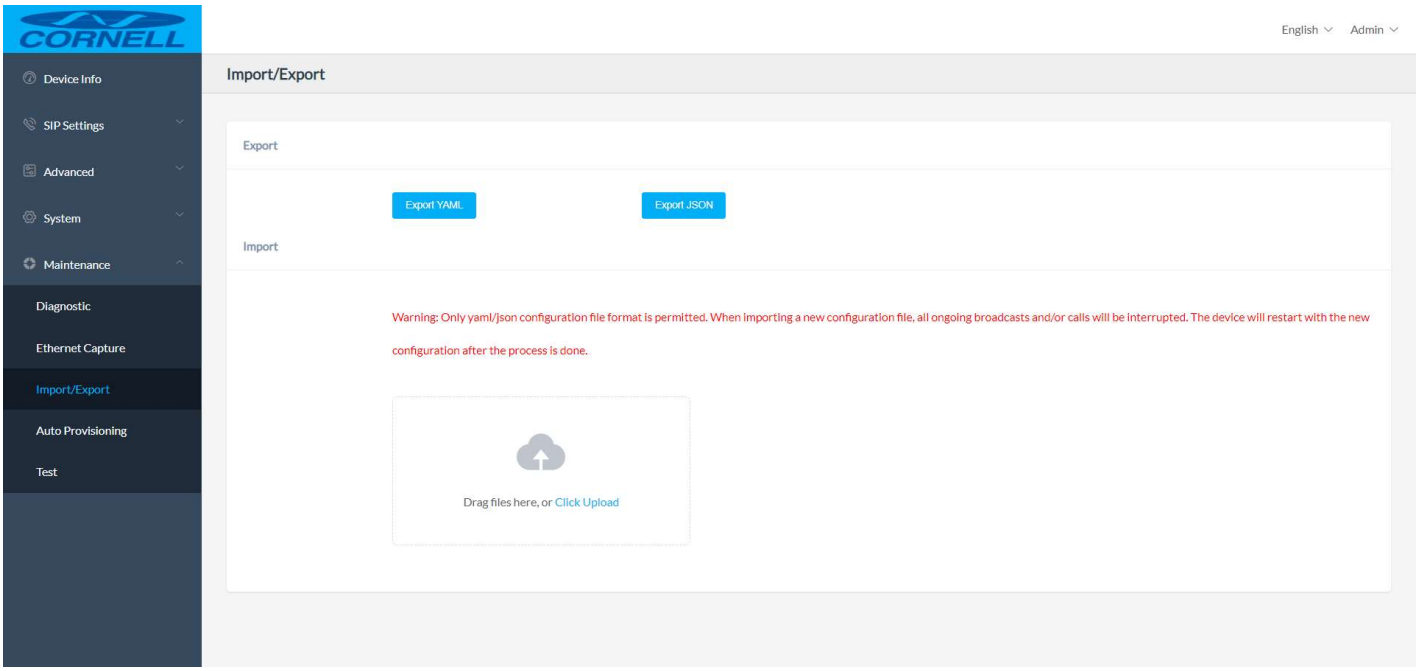
Ethernet Capture

The Ethernet Capture tool can be used to capture network packets and store them in a standard Wireshark compatible packet capture ".pacp" file to view and analyze the data.



Import/Export

This page is used to import and/or export the current configuration of the device. You may use this configuration file to backup or for recovery of the device settings. Both YAML and JSON formats are supported.

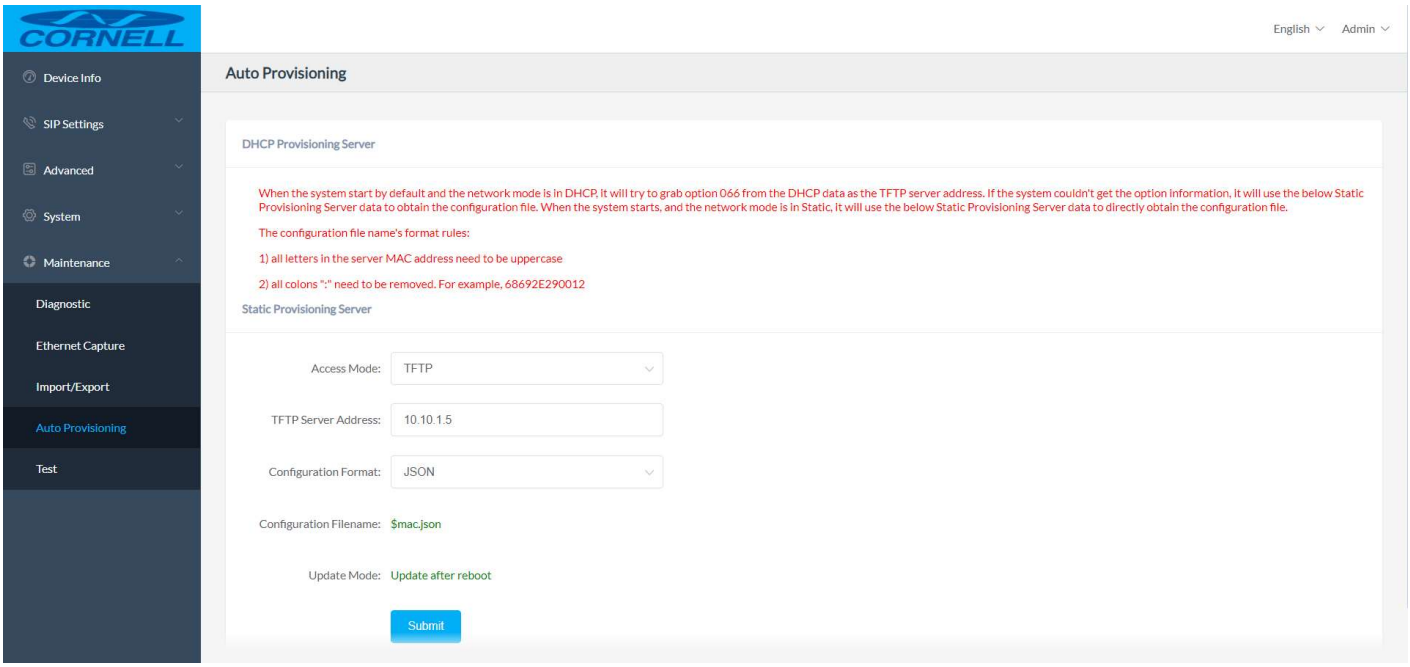


Auto Provisioning

The 4800VS-3 device supports two auto provisioning methods, DHCP Option 066 and TFTP/HTTP. If the network mode is set to DHCP, the device will try to grab option 066 from the DHCP data as the TFTP server address. If the device could not obtain the information, it will use the Static Provisioning Server data to obtain the configuration file. When the network mode is set to Static, the device will use the Static Provisioning Server data to directly obtain the configuration file / settings.

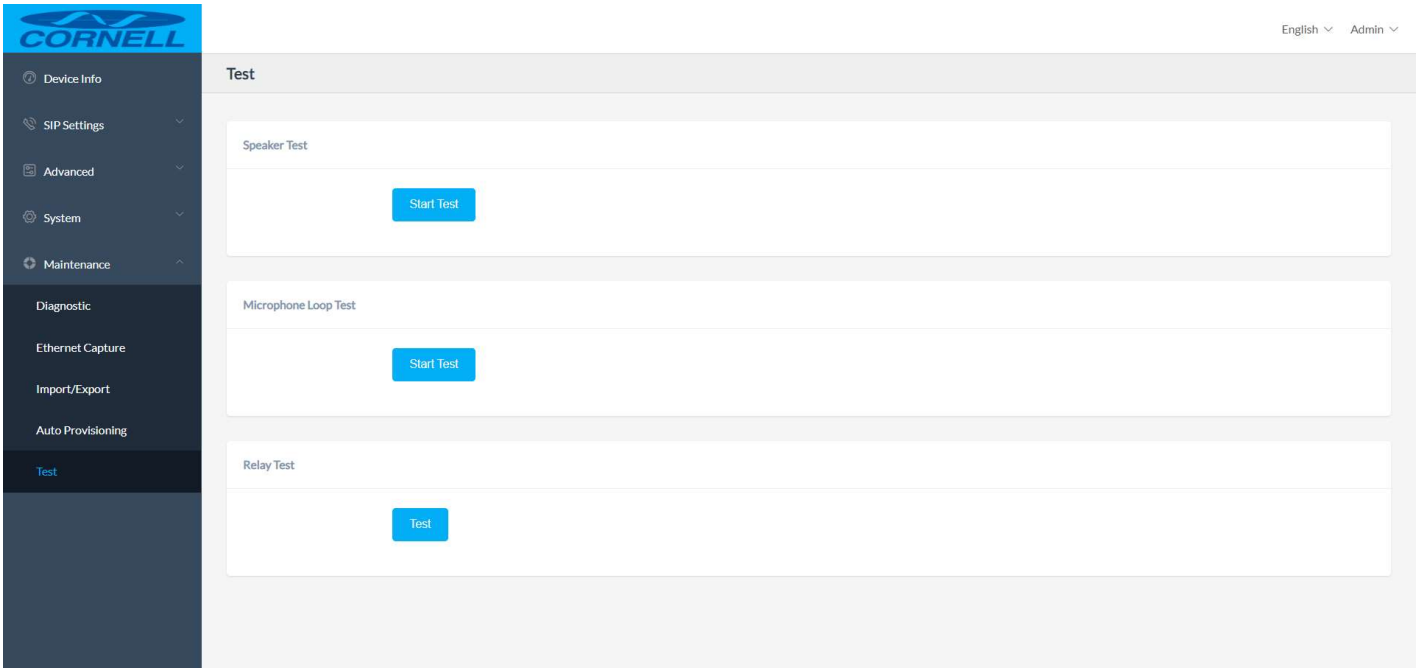
There are two rules for the configuration file name's format:

1. All letters in the server MAC address must be uppercase.
2. All colons ":" must be removed from the file name.



- **Static Provisioning Server**
  - **Access Mode:** default is TFTP
  - **TFTP Server Address:** enter the TFTP server address to be used for provisioning
  - **Configuration Format:** default is JSON / .json file type.
  - **Configuration Filename:** see the name of the .json configuration file
  - **Update Mode:** see when the device will attempt to load configuration / auto provision

Test



- **Test**
  - **Speaker Test:** Clicking the Start button causes the speaker to play a ringtone to test if it is working correctly.
  - **Microphone Loop Test:** Click the Start button, then start speaking to the device. If the microphone is working correctly, you should hear the voice back.
  - **Relay Test:** Clicking the Test button causes the relay device to activate to test if it is working correctly.